Comparison of Several Methods for Separation of Harmonic and Noise Components of Musical Instrument Sound

Ondřej Moravec

Faculty of Music of the Academy of Performing Arts in Prague
Malostranské nám. 13, Prague, CZECH REPUBLIC
moravec@hamu.cz

Abstract
The sound of musical instruments consists of harmonic and noise components. Several separation methods were developed for the purpose of individual analysis of these components. In this paper the individual methods are described, their properties are compared and applicability for various classes of signals is discussed. The behaviour of separation methods is demonstrated with examples of violin tones played by musician.

Introduction

Noise components of musical instrument sound may have an influence on tone timbre and on the perceived sound quality of the tone. For example when the tone D6 (f_0 = 1175 Hz) is played on the violin, the important noise components fall into the interval 230 ÷ 800 Hz, i. e. below the fundamental harmonic [Štěpánek, Otčenášek (1999)]. When the low tones are played, these noise components fall between the harmonics. For the purpose of individual analysis of the noise part and the harmonic part of the tone the separation of these parts is needed. This paper describes several approaches to the task and discusses their respective benefits and disadvantages.

Experiments

Methods of separation of harmonic and noise parts of instrument sound are based on the tone model described e. g. in Serra (1997), which assumes the tone as a combination of the set of harmonics (sinusoids with certain amplitudes and frequencies) and the noise part

$$s(t) + \sum_{i=1}^{N} A_i * \sin(\omega_i(t)t + \varphi_i) + n(t)$$

where $A_i(t)$, $\omega_i(t)$ may be constants or time variables.

One method working in the time domain have been developed and tested. This method consists in finding parameter values of the individual harmonics and successive resynthesis of replica of the harmonic part of the tone. In the first phase the processed signal is partitioned into M segments about 20 ms long. The FFT is applied on each segment and the instantaneous frequency spectrum is computed. Local maxima representing individual harmonics are identified in each spectrum. The instantaneous parameter values (A_{ij}, ω_{ij}, φ_{ij}, $i = 1..N$, $j = 1..M$) are computed and stored in memory. In the second phase stored parameter values are used for additive synthesis of the signal which represents harmonic part of the tone without noise. If the replica is the perfect representation of harmonic part, its subtraction from the original signal gives the noise part of the tone.

Other methods are based on common signal filtration. Depending on which tone part we need to extract from the signal, we require a filter bank consisting of a bandpass (for extraction of the harmonic part) or a bandstop filters (for extraction of the noise part). Methods differ according to how the filter bank is designed (comb filter or cascaded band filters) and by algorithms of filter design.

Results and discussion

The first separation method mentioned above uses the McAuly – Quatieri analysis / synthesis algorithm [McAuly, Quatieri, (1986)]. However, it has some limitations, which disqualifies it for extraction of the noise part. The main limitation consists of constrained frequency resolution defined by the sampling frequency and the FFT window length. Under certain premises the estimation of frequency of the individual harmonic may be improved by way of cubic interpolation of frequency spectrum in the vicinity of the expected frequency. The length of signal segment also defines the methods time resolution. Due to relatively long segments the speed of detectable signal parameter changes is relatively low, and the faster ones are averaged over the segments. Due to these limitations phase difference occurs among the replica and the true harmonic part of the signal so the subtraction of the replica from the original signal did not results in the harmonic part cancellation.

Separation of the harmonic and tone parts by filtering consists of two phases - filter design and signal filtration. The requirement of maximum preservation of the signal properties results in linear phase FIR filter design. The separation of the harmonic and noise parts makes greaters demands on the filters; passband attenuation should be as low as possible and constant over all relevant frequencies, so that amplitude distortion is avoided. The stopband attenuation should be relatively high becouse of the differences among amplitudes of the harmonic and amplitude of the noise part may be more then 50 dB. Each bandpass / bandstop filter should be very narrow so that the separation does not affect the noise part of the signal. These requirements result in the application of high order filters. On the other hand, reasonable computing time during computation of filter coefficients limits the filter order we can use in practice.

The task has two basic solutions – a filter bank of bandpass filters in the case of harmonic part extraction, or a filter bank of bandstop filters in the case of noise part extraction. The bandpass filter bank may be implemented by the two different ways:

- a single bandpass comb filter
- a parallel set of simple bandpass filters tuned to frequencies of the harmonics.

The bandstop filter bank may be implemented by means of dual structures:

- a single bandstop comb filter
- a cascade of simple bandstop filters tuned to frequencies of the harmonics.

The cascade / parallel filtering make more demands on filter behavior in the passband since the attenuation variations cumulates during repetitive filtration.

The signal processing and filter design was done in a MATLAB by Mathworks Inc. using standard functions of its Signal Processing Toolbox.

The functions for design of filter with arbitrary frequency response were used in the comb filter design. MATLAB incorporates several functions which iteratively approximates a given ideal frequency response using different algorithms and different criteria for terminating the process.

The function ***firls*** uses the algorithm described in [Parks, Burrus (1987)]. It designs a linear-phase FIR filter that minimizes the weighted, integrated squared error between an ideal piecewise linear function and the magnitude frequency response of the filter over a set of desired frequency bands.

The function ***fircls*** designs a linear-phase FIR filter that minimizes the weighted, integrated squared error between an ideal piecewise linear function and the frequqncy response of the filter, and restricts the frequency response values to given constraints. The algorithm was published in [Selesnick, Lang, Burrus (1996)].

The function ***remez*** designs a linear-phase FIR filter using the Parks-McClellan algorithm [IEEE Press (1979a)]. This algorithm uses the Remez exchange algorithm and Chebyshev approximation theory to design filters with an optimal fit between the desired and actual frequency responses. The filters are optimal in the sense that the maximum error between the desired frequency response and the actual frequency response is minimized. Filters designed this way exhibit an equiripple behavior in its frequency response.

Comb filters seems to have some advantages: once the filter coefficients are computed, the frequency response of the filter can be easily displayed and checked. The signal filtration can also be done in one pass (in the ideal case). However these filters have some important disadvantages. There is not full control over the design process. The filters exhibit ripples both in passbands and stopbands. Especially the passband ripples (more then 6 dB in some cases) cause distortion during filtration of real musical signals. These signals often exhibit frequency modulation which is converted to amplitude modulation on ripples. In some cases the resulting filter response may be far away from the desired one. Despite of having a high order the filters do not have sufficient stopband attenuation to supress the harmonics, and the filtratiom should be done in two or more passes. Finally the design of high order filters with complicated frequency response is a very time-consuming process.

The application of cascaded / parallel connected simple bandstop / bandpass filters proved to be a more efficient solution. It consists in multiple application of the ***fir1*** function built in the Signal Processing Toolbox. The window-based FIR filter design algorithm implemented in the function is described in [IEEE Press (1979b)]. The final frequency response of the designed bandpass filter is monotonic in the passband and have ripples in the stopband. The disadvantage of multiple filter design and signal filtration is compensated by relatively fast filter design (even for a filter order ≈ 8000).

We considered tones without variable pitch in the extraction task discussed above which lead to our using of a steady filter(s) for filtration of the whole tone. However, many tones played on musical instruments exhibit pitch changes over the time course, mainly in attack and decay parts. As the filter cannot have an infinitely narrow band, small pitch deviations are covered by the filter bandwidth. Pitch changes larger then half the filter bandwidth may degrade performance of the steady filter separation method as can be seen in Fig.1, and require in these cases wider filter bands or adaptive filtering. Too wide filter bands may also degrade the separation process - extracted harmonics may be affected by noise, or large „holes" may arise in the noise part spectrum. The simple adaptive algorithm consists of segmentation of the signal, individual computation of the filter coefficients for each segment and successive filtering of each signal segments using appropriate filters. This process requires multiple computation time in comparison with single filtering. In some cases an audible changes of signal may arise on the segment borders.

Conclusions

Several methods of the extraction of the harmonic and noise part of the sound of musical instruments were discussed. The method based on the resynthesis of the harmonic tone part exhibits relatively good properties for harmonic tone part modelling, but it is not sufficient for noise part extraction by subtracting the harmonic part replica from the original signal. The methods based on filtration of unwanted componens show good properties on signals with a constant pitch. The methods based on multiple filters structures show a better performance then methods based on a single filter with arbitrary frequency response. In some cases the piecewise processing of tones with large pitch changes exhibits audible signal changes at the borders of signal segments.

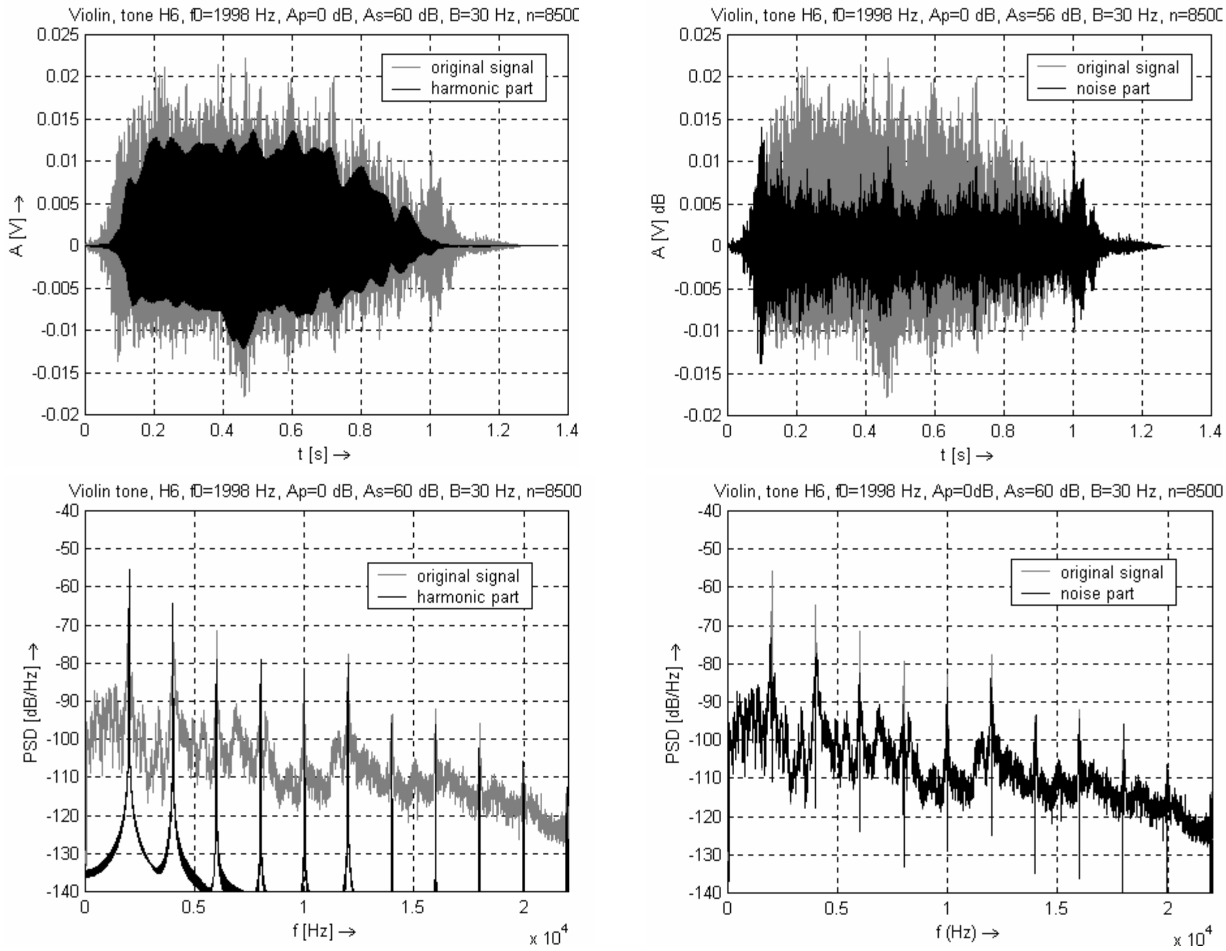

Fig. 1: Example of harmonic (left) and noise (right) tone extraction.Time courses (up) and spectra (down) of separated tone parts compared to original signal. Violin tone H6 (1976 Hz), separation with parallel set / cascade of steady bandpass / bandstop filters.

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